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Research Department Report

HDTV SOUND:
Programme production developments

D.J. Meares, B.Sc., C.Eng., F.I.O.A., M.I.E.E.

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

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Summary

The success or otherwise of multichannel sound with HDTV will depend to great extent on the programme makers being able sensibly to exploit the sound system in order to enhance the listening/viewing experience. This Report documents the BBC's programme experiments over a period of about two years and presents the conclusions of that work.

Index terms: *Sound; multichannel systems; HDTV*

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Ian Childs

Head of Research Department

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1. AN OVERVIEW

If HDTV sound is to introduce multichannel/ surround sound presentations, multiple languages and other options[1,2,3] into the television environment, then it will inevitably make an impact on the normal way of working for television programme companies. It will be shown that that impact need not necessarily be too large but there will be areas where changes are needed.

In the studio, very few programmes currently rely on a single mono or stereo microphone[4]. Presenters and panellists would normally be provided with a personal microphone each, while orchestras are normally recorded with a multiplicity of spot microphones, coincident pair main microphones and ambience microphones. Different mixing techniques have already been shown to be capable of taking these existing microphone arrangements and producing acceptable surround sound programme balances[5]. As will be seen later, however, slightly different microphone practices can make better surround presentations.

The sound mixing room is where most of the change may well be required, but those changes are likely to be concentrated in a few additional or alternative facilities rather than mammoth global changes. It will be seen that new sound panning facilities will be needed on the sound desk, as well as changes to the monitoring arrangements. Equally, it should be stated that many programmes have already been produced with virtually no changes to the existing stereo facilities. So the extent of the changes could be based on the extent of penetration of the new services into the market.

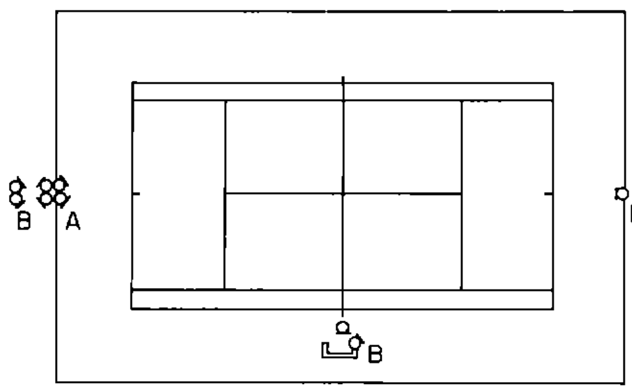
The only major production commitment that needs to be changed if these new services are to be introduced, is the number of sound channels to be recorded and routed with the video signal. With the ingress of digital audio into the fabric of the broadcasting infrastructure[6], multiplexed digital audio routeing will be little or no problem, but there will always be the problem of recording the extra sound signals, until HDTV recorders with extra sound tracks become the norm.

2. ON-SITE RECORDING

The on-site recording problems vary enormously depending on the venue. Apart from the 'routine' problems of setting up the recording infrastructure, the success or otherwise of the programmes has centred around having sufficient microphones in the right places to enable the post-production mixing to take place. This has also determined whether real or artificial 'sound fields' can be created in the listening environment.

The examples given below, whilst not recorded solely for a single specific surround sound format, were mixed initially to a 3/2 format. Whilst other formats have been assessed, the production comments below, except where stated, are based mainly on the experience of the 3/2 presentation, which is claimed by many to have most potential without making excessive demands on producers or consumers.

Recordings at the Wimbledon Tennis Championships in 1989 and 1990 were planned from the start to present a sound picture that was as simple as possible. The concept was that, as the pictures would concentrate on one basic view of the Centre Court, i.e. down the centre line of the court, the sound impression should try to match this as closely as possible. To this end a sound field microphone (SFM), placed on the centre line at the boundary between court and spectators' stand, was used as the main feed, with spot microphones (mostly gun microphones) used as fillers: Fig. 1 shows the arrangement used. Even with this venue, time delays across the court were problematic and the single gun microphone placed at the far end of the court had too great a delay to be used; for far-court effects the gun microphone on the umpire's chair was found to be sufficient. (In this situation of widely-spaced moving sound sources and



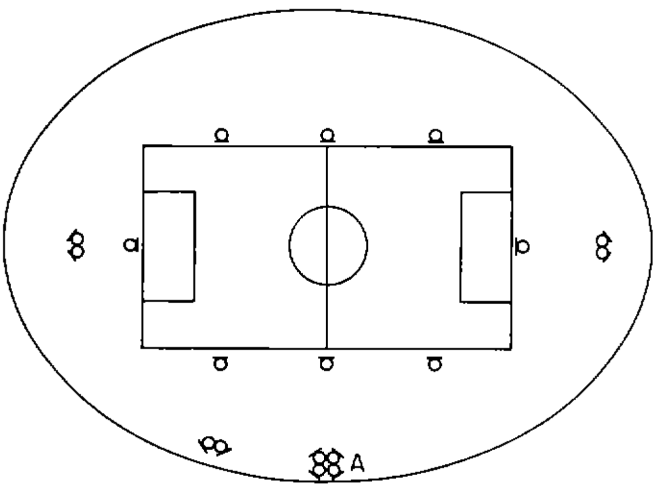
A = sound field microphone
B = gun microphone, Sennheiser 416 and 816

Fig. 1 - Wimbledon: Centre Court.

distant gun microphones, it was not possible to prescribe appropriate compensatory audio delays, as would have been the practice for orchestral recordings.) The mix obtained was intimate and most enjoyable and, perhaps because of the inter-channel relationships generated by the simple microphone arrangement, was relatively easy to matrix down from surround to three channel and stereo reproduction formats.

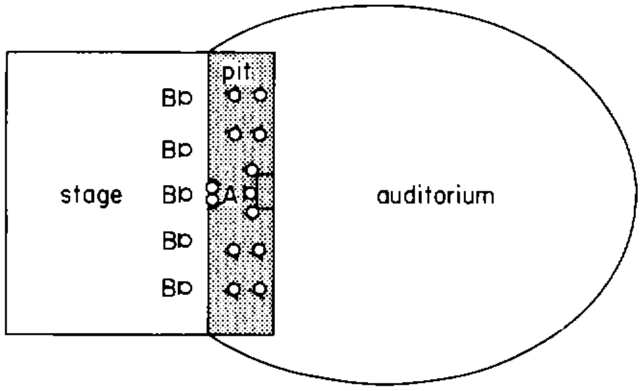
The FA Cup Final at Wembley 1989 was a much more difficult environment to record in. It is vast by comparison to the Centre Court; it has a huge crowd of 82,000, giving rise to substantial swings in both sound level and sound location as first one section of the crowd reacts and then another. Amidst all this, the recording engineer still has to try to pick up the sound of the ball being kicked somewhere on the pitch. Attempts were made to deduce in advance a good location for the SFM, bearing in mind that nothing could be suspended above the pitch. (See Fig. 2.) In the event, the SFM output was so heavily biased towards the nearest section of the crowd that it was unusable. For this recording, gun microphones on both the crowd and the pitch effects were used exclusively. The aim at the mix down was to create an exciting though, admittedly, 'unreal' sound event, by surrounding the listener with crowd effects. This was achieved successfully, though the perspective on the crowd was as though the listener had been reserved a large block of seats entirely to himself, so that no-one else was within several feet of him.

For a recording of 'The Prince of the Pagodas', at The Royal Opera House, Covent Garden in London, the normal stereo microphone arrangement, as shown in Fig. 3, was used. This comprised a



A = sound field microphone
others = Sennheiser 816

Fig. 2 - Wembley Stadium.



A = C422 (main pair)
B = PCC160 (PZM)
Others = mix of U87, KM84, C414, C460 and B & K 4006

Fig. 3 - The Royal Opera House.

combination of a main stereo pair over the proscenium arch, spot microphones on the various sections of the orchestra in the pit and pressure zone microphones on the front edge of the stage. The principal aim was to capture the orchestral layout using the stereo pair, supplementing this with spot microphones and stage microphones as necessary. For this venue it is not acceptable to have front-of-house microphones, other than the discreetly placed stereo pair. It was the intention of the mixing session to simulate the ambience, using artificial reverberation fed with the existing microphone feeds; this was probably the best arrangement for other reasons, as the hall was relatively dry and front-of-house microphones would inevitably have picked up audience noises (see later under Royal Albert Hall). The main orchestral acoustic picture was given by the stereo microphones panned three quarters left of centre and three quarters right of centre. (Though this did not present any great problems, it was felt, at the time of the remix, that a triple microphone arrangement specifically for the three front channels, or a SFM suitably decoded, would have been better.) To this was added sufficient signal from the spot microphones in the orchestra pit, to enhance the presence of the orchestral sounds. The stage microphones were not ultimately required as there were sufficient stage effects on the main microphones. In generating the rear channel feeds, additional delay was introduced into the main microphone feeds and artificial reverberation, fed from a mix of the spot microphones, was added. This produced a rather subtle surround mix, although not one totally free of delay effects. (Better use of compensatory delays, in the initial mix fed to the reverberator, may have avoided this.) One other point, discovered during the mixing, was that a shortage of tracks (only 24!!) at the recording venue had led to sub-mixes being recorded for a number of orchestral

sections. These were normal stereo sub-mixes, whereas a three channel sub-mix would have been more useful in post-production.

Several recordings have now been made at the Royal Albert Hall with improvements taking place on each successive visit. The last one was the most ambitious in its post-production intentions and therefore the most revealing as a pointer to the future. As shown in Fig. 4, a multiplicity of microphones was used for the sound pick up, with equalising delays imposed on the spot microphones to effect approximate co-timing. The main orchestral 'picture' was created by suitable panning of the five omnidirectional microphones suspended in a curtain over the front ranks of the orchestra. Spot microphones were mixed into this frontal presentation to enhance contributions from the more distant orchestral sections. Finally, the hall microphones were panned to the sides and rear to create the surround sound. The mix created a well defined frontal sound stage, with natural reverberation mixed to create a realistic sound field. Not only were sources found to feed the rear channels in isolation, but additional feeds, appropriately timed, were mixed half way down each side of the listening area to fill in the hole-at-the-sides. In surround sound this worked extremely well, giving a very real impression. The problem was however that it made very poorly-compatible stereo and three channel presentations, simply because of the amount of real audience noise that was then overlaid on top of the orchestra. The only solution, using the available prerecorded tracks, was to create an artificial ambience, as with the Covent Garden recording, using the orchestral microphone signals to feed the artificial reverberator. The ultimate for such a recording would be to use a combination of the two, with real ambience being used as much as possible, e.g. before and after the music, and artificial ambience during the concert when audience noises become disturbing. This would also enable full exploitation of the real hall sounds on those occasions where the Promenaders join in.

One final production experiment which is worthy of note was a joint BBC/IRT/WDR experiment in Köln*. This used, as its source, a multitrack recording of the BBC programme Horse of the Year Show, which covers a group of show jumping competitions. This was interesting, primarily because the recording was made with no thought of surround sound; the microphones were rigged entirely for the transmission of stereo programmes. The layout of microphones at the stadium is shown in Fig. 5, with eight gun microphones suspended high above the arena and additional stereo microphones directed at one section of the audience. The intention of the

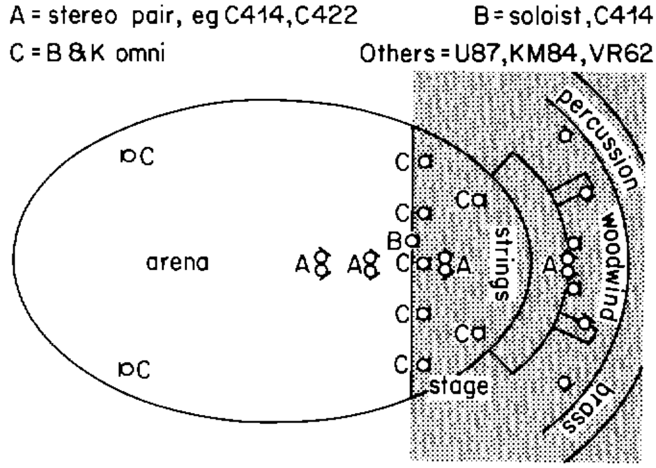


Fig. 4 - Royal Albert Hall.

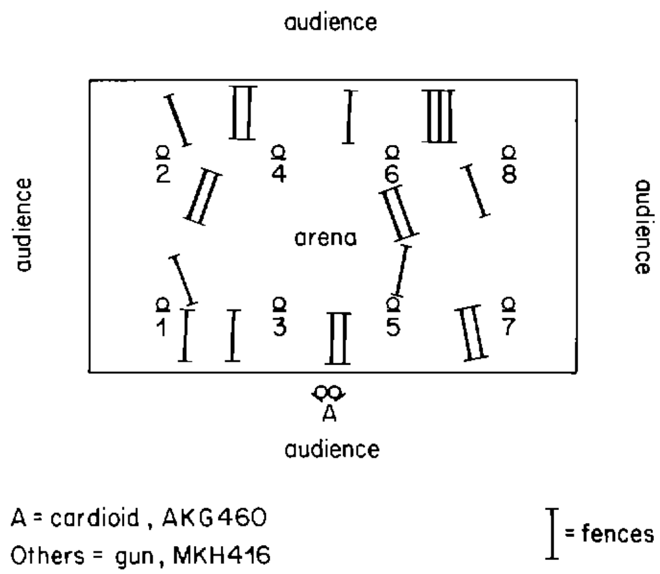


Fig. 5 - Horse of the Year Show.

programme mix was to present a stable wide frontal presentation of the action in the arena, with crowd noises and reactions coming from all round the listener. There were however, two problems in carrying this out. Firstly, there was so much sound energy in the arena microphone signals, coming from the stadium public address and crowd, that a great deal of gain riding of the arena microphones was necessary in order to pick out the horse and rider effects. This is however normal for a stereo production. The second point is one where much more work is required to find an optimum solution, and that is the normal tendency to cut between vastly different camera positions to give the optimum visual presentation whilst, of necessity, retaining a fixed aural perspective and view of the events. These points apart, the sound presentation was a success, despite the absence of any advance planning for the surround sound.

* This recording and mixing experiment is the subject of further discussion in the companion Report[1].

This last programme, and an orchestral recording by WDR, were used during the same experiments to enable different sound presentations and number of channels to be compared. Of particular importance were the separate comparisons of front channel presentations and rear channel presentations. In brief, the general conclusion was that there was a law of diminishing returns applying to both. When properly mixed/balanced sounds were used, three front channels/loudspeakers gave a much more enjoyable presentation than two, but four was not a great improvement on three. Similarly for the surround sounds, a single additional channel gave much worse realism than two, but four was not significantly better than two. This certainly supports the BBC's previous findings on reproduction formats.

3. POST-PRODUCTION ACTIVITIES

Whilst it can be concluded therefore that for many productions the sound pick-up problems of surround sound can be overcome, what about the potential difficulties in the control rooms and the need for changes in techniques and technology?

Two post-production environments have been used extensively by the BBC for its sound mixing experiments: they are Sypher 2 and the control room of the Music Studio (TMS), both at BBC Television Centre in London, see Figs. 6, 7 and 8. Both areas are relatively cramped, particularly for surround sound; both areas are full of technical equipment and need careful rigging of the stereo facilities in order to make the multichannel mixes. However, it must be borne in mind that both areas are in many ways typical of television sound areas, except that they are larger than most, at least in the BBC. This is a significant point to remember when deciding how HDTV programmes will be balanced in the future. There are probably very few broadcast studios or outside broadcast facilities that could easily accommodate multichannel or surround sound monitoring, without at least a degree of reorganisation. (By way of contrast, see Ref. 7.)

However, by careful adaptation it has been possible to use both Sypher 2 and TMS successfully, though in each case at least half a day was needed to effect the necessary reorganisation of the stereo facilities. Particularly time consuming, were the rigging and balancing of the loudspeakers and the plugging/switching of the stereo desk routeing system to enable surround sound processing to be achieved.

Absolutely essential in the rigging of the loudspeakers, is a type of loudspeaker that is capable of generating a precise, sharp image when a common signal is applied to a pair of loudspeakers spaced at about 60°. The monitoring room also must be restrained in the effect it has on the sound. In this context, the BBC's tendency to have very heavily treated monitoring rooms (reverberation time $<= 0.2$ sec) has proved to be advantageous. (It is also probably fortunate that the BBC has, in the main, so far avoided following the pop music fashion of Live-End-Dead-End acoustic design, but has instead continued to try to achieve an even distribution of acoustic treatment. Loudspeakers in various parts of the room are therefore likely to interact with the room in a similar way to one another.)

Fig. 6 - Multichannel audio production in Sypher 2.

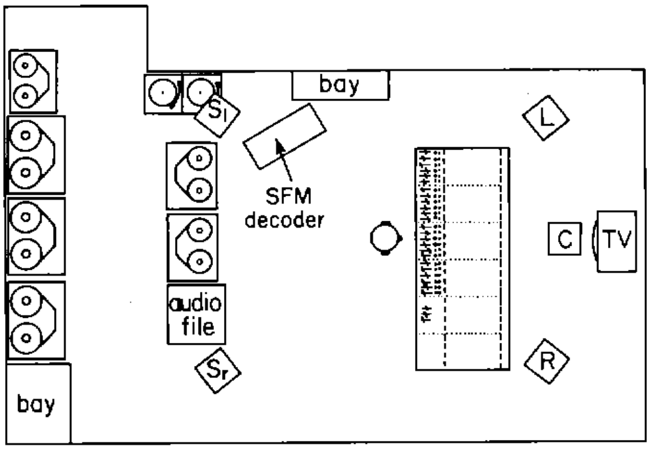


Fig. 7 - Sypher control room layout.

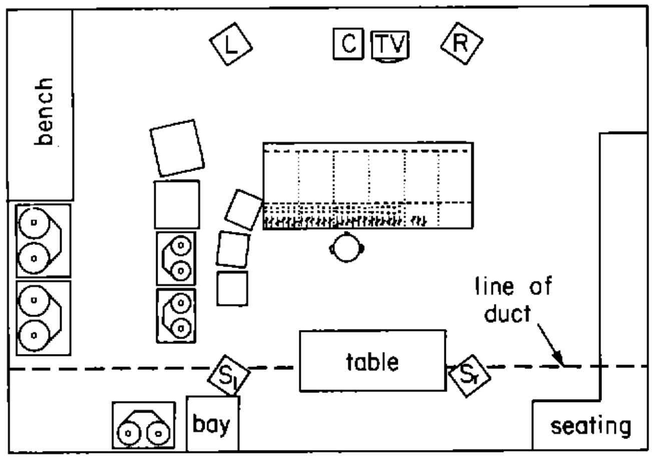


Fig. 8 - Control room of Music Studio.

Once a listening centre has been defined, the loudspeakers should be located as required. In neither of the two rooms used by the BBC so far, was this particularly easy. In the Sypher Suite, a large amount of technical apparatus behind the monitoring position had to be moved back towards the rear wall. Even then, the rear loudspeakers were mounted on a frame over a bank of tape recorders. In the case of TMS, a large encased ventilation duct had been installed along the rear wall below the level of the false ceiling. The rear loudspeakers should ideally have been placed under this, on the basis of distance to the monitoring point, but this would have seriously affected their tonal quality. Two solutions were used to overcome this: on the first occasion, the signals to the rear loudspeakers were delayed to compensate for the distance error, whilst for the last session, all loudspeakers were moved closer to the monitoring position. (A third solution of opening the rear 'stage', such that the rear loudspeakers subtended an angle of about 120 degrees[8], was not possible in either TMS or Sypher because of the location of technical equipment.) The final problem in the placement of loudspeakers is that ideally one would wish to have the centre front loudspeaker and the picture monitor co-located. For the mixing rooms, the loudspeaker has always been allowed to take pride of place, with the picture offset. For reproduction or demonstration rooms, the picture must be placed centrally with the centre channel either above or below it. Additionally for HDTV CRT displays, care must be taken to screen the display from the stray magnetic flux of the loudspeakers; this may on occasions require the encapsulation of the toroidal magnets in the loudspeakers.

Balancing the levels of the loudspeakers is a straight extension of the technique used for stereo. A common signal is fed in turn to each adjacent pair of loudspeakers and the sensitivity of one of them is adjusted to create a central image. The advantage of the aural approach is that it is far more sensitive (accurate) than the alternative of a sound level meter: furthermore, sound supervisors are used to making aural judgements — they are not necessarily experts in the use of a sound level meter. It should also be noted that identical or closely similar loudspeakers should be used for all channels. It is misleading to use different (cheaper) loudspeakers for the rear channels, particularly at the sound balancing stage, where one may be tempted to alter the balance to compensate for some of the deficiencies of the monitoring system.

Mixing desk routeing has required care, specifically because of the stereo organisation of most sound desks. Several solutions have been found, which may not be elegant but which do provide multichannel routeing on a stereo desk. Two other points are, however, worth further discussion here, namely automated mixing and panning. Experience would now indicate that automation is absolutely essential for post-production editing of surround sound. Considering the task of editing, one has somehow to carry the surround effect across each edit point with, on most occasions, no significant change at the edit point. For sports this is far from easy, simply because of the dramatic changes in crowd noise that are likely to occur at an edit. The solution is to fade-in the post-edit sound well before the edit and fade-out the pre-edit sound well after the edit; but even then, each edit may need several attempts before the effect is acceptable. With the number of channels involved, manual control of the whole event is virtually impossible, even for a skilled operator.

Panning also requires great care if one is to use the three frontal channels and the surround channels to their best. The most clearly defined sound images* are those created by feeding audio signals to adjacent

* In the author's opinion this is still a valid statement, but suggestions have been made that the use of all three front loudspeakers could create better panned images. Whilst this may well even out any variations in the quality of the images across the stage, it has still to be proven that sharper images can be created.

loudspeakers. Thus to pan a sound from left to right, the sound must be cross-faded via the centre channel. Stereo desks are not organised to operate this way but can be forced to provide such a facility for non-varying, static location, panning. Dynamic panning will require some changes to the format of a normal stereo desk, and several workers in this field are already studying the problem. Not only has the principle of multichannel panning to be accommodated, but it is necessary to determine the optimum panning law. Stereo panning has usually been provided by means of a sine/cosine relationship between the two channels, but there are already strong arguments[9] for different panning laws for surround sound and even laws involving more than two channels at once. However, it is predicted that, in the early stages of multichannel operations, at least, revised dynamic panning will be provided by means of an add-on box to a standard desk in order to minimise the financial impact.

The other area of mixing and post-production where stereo devices can be adapted for multichannel working, is that of sound effects. To date, only stereo devices have been available and therefore there was no choice. Experience has shown however that multichannel solutions would be better and easier to use. This comment applies not only to DSP devices such as artificial reverberators, delays, flangers, etc., but also to sound effects discs. The opening section of a tennis programme which the BBC produced, used three stereo sound effects discs to create the multidimensional ambience that was needed to accompany a high crane-mounted camera shot of the tennis club and its environment. Surround sound recordings would obviously have simplified the production.

One essential change of facilities that will be needed, both for production and post-production areas, is the provision of multi-format monitor switching[10]. Current stereo productions include occasional switches to monophonic monitoring for assessment of compatibility. Likewise the surround sound production will need to be monitored for compatible reproduction in stereo, three channel and any other formats that are considered to be representative of a reasonable proportion of the systems being used by the audience. Current methods of providing for this multi-format switching are cumbersome or expensive or both, even though the technology is in principle relatively simple. In the near future this will be most easily provided by an add-on box placed in the signal feeds between the desk output and the loudspeakers. In the longer term, it could become a standard extension of the monitor switching already provided in the sound desk.

As already mentioned above, the other development that will ease the introduction of multichannel sound into the production environment, is the provision of HDTV recorders with multiple sound channels. The EBU has already recommended to various groups that such a recorder should have at least eight channels of audio, with more for those machines required to provide for track-bouncing or assembly editing. If radical increases in track numbers are required, or where track-laying for post-production work is being planned, then a separate synchronised audio recorder will continue to be used as now.

4. EXPLOITATION OF MULTICHANNEL SOUND

A great deal of experience by many broadcasters with Television Stereo, and a more limited amount of experience with multichannel sound for television, has led to an expectation of what can be achieved in the new era of HDTV Sound. To some extent it benefits from and matches the experience of the cinema industry, but in at least one specific aspect the conclusions differ.

In the cinema industry, the programme maker has to cope with an extreme range of reproduction environments and listener placements. The cinema can vary from a small intimate 20/50 seat environment with very low reverberation, to a large auditorium seating many hundreds of people with a comparatively long reverberation. The viewing/listening angle can vary over almost 180 degrees, and whatever the conditions, the producer has to try to generate spatial coincidence between the sound image and the visual image. This has led[11] to the use of the centre channel in the cinema sound system for dialogue, with virtually no dialogue from any other location. In the television industry however, the normal reproduction environment domestically will be much smaller, more intimate and less reverberant. The listening position will also be much less extreme than the worst positions in the cinema. This opens the horizons for the programme maker, who can at last contemplate the use of directional cues on the voices to add a new dimension to the production. This is already being seen as being valuable in television stereo and, indeed, some IMAX productions[12] and experiments into systems offering 'virtual reality'[13] are exploiting spatial sound techniques in an attempt to get closer to the real experience. Whilst 'virtual reality' will be a long time in coming to HDTV, the more limited spatial representation of dialogue will be seen to be increasingly important in HDTV programme production. (See also Fig. 9.)

The way in which film sound tracks will be handled is itself a debatable matter. If surround sound encoding has already been implemented, say using the

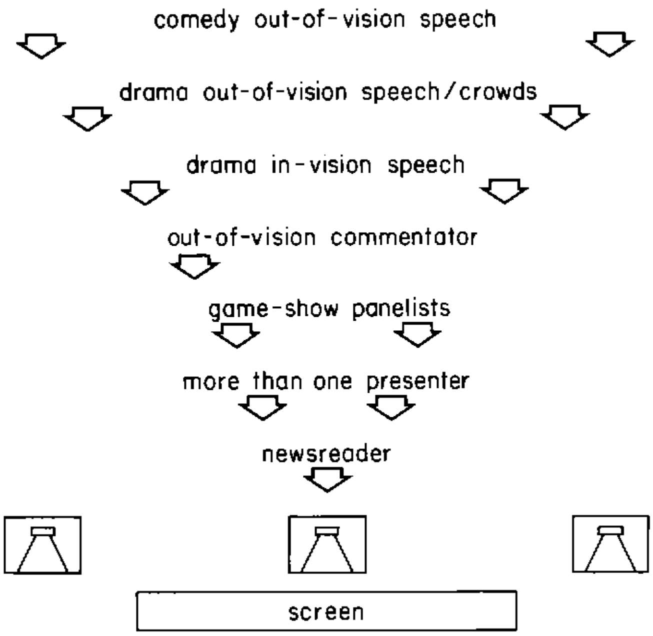


Fig. 9 - Distribution of sound images.

Dolby encoding format, why not just radiate those signals as they are? It should be remembered however that Dolby Surround is just one of a number of formats that exists already and from which the HDTV programmes will be sourced. There are also additional formats being developed and marketed[14]. These various film formats have to be brought to a common transmission format prior to broadcasting, if the reception decoding problems are to be reduced to sensible proportions. Additionally, it should be borne in mind that the film sound tracks are mixed to be compatible with a very wide picture and when the picture is reduced, even to HDTV size, there are going to be occasions where the sound stage, particularly for in-vision effects, is going to be too wide. By decoding the film sound formats to a common 'loudpeaker-feed' format at the studio centre, not only can the best possible decoder be used in each case, at no cost to the end customer, but also, the sound presentation can be modified if necessary to suit the somewhat smaller screen. It is because of such optimisation matters, that the CCIR and EBU groups studying HDTV sound are recommending members to decode film sound tracks prior to transmission.

It will be important in other ways as well, not to over-stretch the brain's desire to have sounds coming from all around the head. In real life one has the ability, when one's interest is aroused, to turn to face an event in order to apply one's sight to the interpretation of that event. If in surround sound such interest is generated, the turn of the head will only bring an empty wall into view. In this context, the sound system must enhance the visual imagery rather than take precedence, and whilst transient sources of sound could be located outside the visual image, such areas should normally be restricted to ambient sounds, see Fig. 10.

Surrounding sounds have, however, always been found to be beneficial when correctly mixed, although this comment refers mainly to systems with two channels of surround. For most of the time, a single channel of surround, whether fed to one or more loudspeakers, is perceived as an independent source with a precise location of its own. This is due to a combination of the spatial angular displacement between this loudspeaker(s) and the front ones, and the different directional properties of the hearing system for sources from the rear. Particularly with crowd noises, the impression given is that there is a large number of people all in one spot, rather than spread across the rear. But it should be noted that 'more is not necessarily better'. Some experimenters, whilst stating that 'a minimum of two (loudspeakers) is needed', found that four gave problems with the timbre of the surround sound[15]. Thus, at least two channels of surround sound will be needed, but the extent of the benefit of the full surround sound presentation over the multichannel frontal presentation will always depend on the skills of the programme maker and on the type of programme[16, 17].

As the extent of HDTV coverage increases, or even to ensure that it does, there will be an increasing demand for multiple language options. For major sports events (witness the Eureka 95 HDTV relays of the 1990 World Cup football matches and the plans for the 1992 Olympic Games) it will be essential to provide multiple language commentary facilities. This has in any case been foreseen at the planning stages of such systems as D-MAC. There are arguably also strong cases for the provision of at least one extra language for films and co-productions, viz. the original

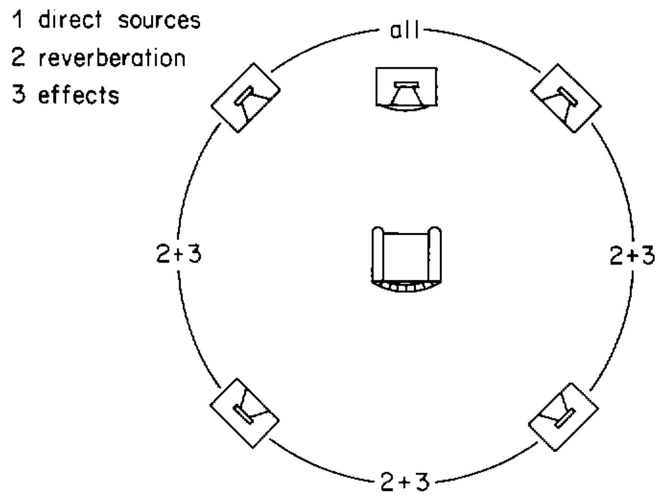


Fig. 10 - Preferred location of sound sources.

Table 1: Estimates of the number of adults in Great Britain arranged into different hearing loss categories.*

Description of hearing loss (BSA categories)	dBHL better ear average	Number of adults (millions)	% of total adult population
Mild	25-40	5.0	11.33
Moderate	41-70	2.2	4.99
Severe	71-95	0.24	0.54
Profound	96 +	0.06	0.14
Total		7.5	17.00

* Based on data provided by the Royal National Institute for the Deaf and the Institute of Hearing Research.

language and an over-dubbed language for the majority of listeners in the particular country of replay. Such provisions are relatively simple for sports, where the commentators are an adjunct to the event rather than part of the action being reproduced. The consequence for sports is one extra channel per language, even if surround sound is being radiated. In virtually all other types of programme, i.e. for all dialogue other than commentary, the production is attempting to place the spoken word into the acoustic environment of the programme. This requires a full mix for each language; three languages in stereo would require six sound channels; two languages in surround sound would require ten channels, etc. Thus multiple language working might in principle be a worthy cause, but it could be expensive for many programmes.

Additional services, such as clean dialogue for the Hard of Hearing (HoH) and dynamic range control, should also be seen as both potential improvements in HDTV sound and potential consumers of the data capacity that is available in HDTV for sound applications. The HoH channel is however one that should be very carefully considered[18]. It is estimated that upwards of 17% of the population of such countries as the UK have some loss of hearing (see Table 1) and that with age the relative percentage increases. It should also be recognised that with hearing loss comes the problem of dynamic range accommodation and difficulties in distinguishing between wanted and unwanted sound in a complex mix[19]. If capacity can be made available for a HoH sound channel (ideally, by reallocating a commentary channel for this purpose†), then the otherwise disenfranchised HoH listeners would benefit enormously. The cost for the programme maker would be the cost of providing clean dialogue (it already exists in the early stages of many programme mixes) and of conveying it to the transmitter.

5. COMPATIBILITY ISSUES

As mentioned above, the artistic requirements of compatibility frequently require changes to be made to the surround sound mix, specifically because of the sound balances achieved in three channel or stereo presentations. The concert recording, above, though good in surround with the audience noise, had to be changed because of the stereo reproduction. Initial programme balances for football and tennis have had to be modified because of image location problems in stereo. The amount of sound energy, e.g. crowd noise, in the surround channels has to be tempered, on occasions, because it would be overpowering in the stereo mix.

But not all programmes need such surround channel attenuation in stereo, even if they can tolerate it. Drama, for instance, may need the full strength of the rear channels to be retained in the compatible presentations. Football crowd noises may need considerable attenuation, whilst concert hall ambience may need very little. There may need to be a mechanism for changing the parameters of the baseband compatibility matrix to suit the programme type, but for obvious operational reasons this should be avoided if at all possible.

To date, various compatibility compromises have been deduced during the mixing sessions, as shown in Table 2. As can be seen, the level changes required for compatible reproduction do vary from one programme type to another but probably not by a

† Whilst the data capacity of a commentary channel is arguably too great for the needs of the hard of hearing channel and the use of sophisticated bit-rate reduction techniques could substantially reduce the data requirements of the channel, the fact remains that there is likely to be much more pressure to standardise on a receiver design that will cope with a commentary channel than there will be pressure for a HoH channel. Thus by adopting a commentary (or language) channel for the HoH channel, the only special feature required by the HoH is a data flag to tell the receiver which commentary channel holds the sound for the HoH. This flag facility is identical in principle to that needed for French, German or other languages. Thus the receiver hardware for the HoH will not be special, and therefore could be expected to be relatively inexpensive.

Table 2: Experimentally deduced attenuations of 3/2 surround sound signals in different forms of sound presentation.

Programme	5-ch to 4-ch	5-ch to 3-ch	5-ch to 2-ch
Football and Tennis	S' = S1+S2−6 dB	L, C, R unchanged S1, S2 unused	L' = L +(C−6 dB)+(S1+S2−9 dB) R' = R +(C−6 dB)+(S1+S2−9 dB)
Promenade Concert 1	S' = S1+S2−x dB x = 6 for audience x = 4 for ambience	C unchanged L' = L +S1 R' = R +S2	L' = L +(C−6 dB)+(S1−6 dB) R' = R +(C−6 dB)+(S2−6 dB)
Promenade Concert 2	Not assessed	C unchanged L' = L +(S1−4 dB) R' = R +(S2−4 dB)	L' = L +(C−3 dB)+(S1−6 dB) R' = R +(C−3 dB)+(S2−6 dB)

very significant amount. It has thus been argued that sound monitoring should encompass all of the various reproduction formats in order to guarantee artistic compatibility of the sound balance.

But such compatibility monitoring cannot be done in isolation from, nor in ignorance of, the transmission system. In the case of normal stereo, a strict transmission relationship exists between the stereo signals and the mono signal; in programme mixing and monitoring that same relationship is used. So it will be for multichannel sound systems of the future; for instance, there will be strict relationships between five channel surround sound, three channel frontal and two channel stereo. To a great extent it will be the artistic compatibility requirements that will determine those relationships, but there may be additional transmission-dependent factors that must also be taken into account. For instance, artistic requirements have already been seen to be satisfied by a compatibility matrix of the form shown in Appendix 1[20, 21]. This, however, requires a reverse matrix in the receiver in order to derive the loudspeaker signals for each form of reproduction. Such matrixing may not be compatible with some of the higher ratios of bit-rate reduction based on psycho-acoustic principles, which are being proposed for transmission in, for instance, DAB and HD-MAC[22, 23, 24]. In this case, alternative proposals for the transmitted signals have been tabled[25]; namely loudspeaker signals for the most complex member of a transmission hierarchy (say 3/2 surround sound), whilst the loudspeaker signals for the lesser complex reproduction formats (say stereo) are derived in the receiver by downmixing[26] (see Appendix 2). The advantage of this, for the programme maker, is that different constraints can be applied to the generation of signals for each form of reproduction (for instance more/less attenuation of the rear sound signals). However, it is obviously even more important in such a system that there be a standard method of downmixing, such that the production staff can be assured that they are hearing the same thing as each of their groups of listeners.

The other aspect of compatibility, required of any transmission format or matrix, is compatibility with different forms of programme origination. Any broadcaster with a library of existing programmes is going to want to be able to continue to exploit that library, regardless of the fact that yesterday's programmes were made with a different sound presentation in mind than that proposed for tomorrow (see Fig. 11). In other words, old programmes with, say, stereo sound have still got to make sense when broadcast through a multichannel sound transmission system. That is one of the additional features that went into the derivation of the compatibility matrix of Appendix 1.

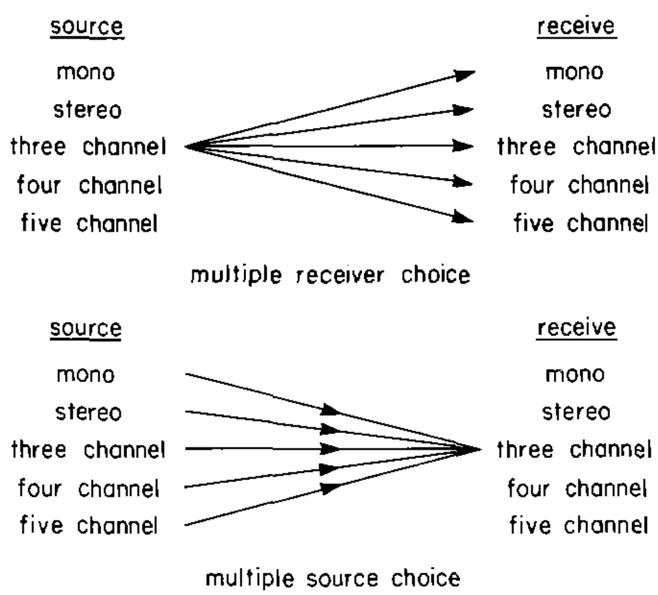


Fig. 11 - The multiple opportunities of a compatibility matrix.

There are also spatial compatibility benefits at the production stage in the use of multiple channels of sound, just as much as for the listener at home. The change from two channels (stereo) to three channels at the front has been found to be particularly worthwhile. As stated for the home listener, the centre channel adds a great deal of stability and sharpness to the frontal sound images for a wide range of listening positions[27, 28, 29]. No longer do the sounds move rapidly to the closer loudspeaker as the listening position moves away from the precise centre line; with three channels the shift in image is dramatically reduced. This is a particular bonus to the sound supervisor, who has, of necessity, to move around by significant distances whilst mixing the programme.

Thus, it can be seen that there is much to be gained by ensuring that the sound system of the future offers compatibility in a number of ways. But that compatibility, as has been seen, makes demands on the transmission system.

6. CONCLUSIONS

Developments in HDTV sound are proceeding apace and various international committees and groups are studying some of the more theoretical aspects of the subject. Equally important, are the experiments reported here on the programme production work, and the multifaceted problem of compatibility. Existing facilities have been shown to be adaptable to the needs of multichannel sound production; but undoubtedly, purpose-designed facilities would be better. The requirements of compatibility with existing audiences always occur with new services, but there are solutions available. Other aspects of the system design, particularly the transmission constraints, will also ultimately influence the development of the compatible multichannel sound system for HDTV.

7. ACKNOWLEDGEMENTS

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APPENDIX 1

Compatibility Matrixing

A1.1 Introduction

Multi-channel sound programmes may need matrixing for two reasons. Firstly, there will be a need to ensure that different modes of reception are possible, regardless of the mode of transmission. Secondly, the transmission bit-rate reduction may itself have need of certain relationships between the various sound channels.

Within the hierarchy of systems being recommended by the CCIR, there are many individual methods of utilizing the sound channels. At the highest level, for cinema types of application, the 4/4 system provides potentially the highest quality of frontal sound image localization and the most natural ambience from the surround channels. At the lower levels, the hierarchy provides for those programme makers and listeners who want to make or listen to stereo or mono sound balances. It is the fundamental aim of a compatibility matrix to provide simple but controlled inter-relationships between these various modes of programme generation and programme reception.

Thus, the matrix has to provide ways in which, say, a 3/2 broadcast can be received in 3/2 format or 3/0 format etc. This is termed '*downward compatibility*', and is discussed in this Appendix.

It also has to enable the archives of existing programmes in stereo to be transmitted over the multi-channel system and to be received sensibly on the various formats of receiver. This is termed '*upwards conversion*' and whilst not specifically discussed in detail any further, provision is made for it in the equations in this Appendix.

Finally, there may be circumstances where intrinsic downward compatibility of the transmitted signals is not required and discrete loudspeaker signals are transmitted, but where, nevertheless, it is required to generate loudspeaker signals at the receiver which relate to a lower order in the hierarchy, i.e. a 3/0 presentation of 3/2 transmitted signals. Under these circumstances, a standardised form of '*downward mixing*' equations will be needed in the receiver: these are given in Appendix 2.

In order to identify specific loudspeakers and channel sources in a variety of arrangements, the codes given in Fig. A1.1 have been used. For the 3/2 presentation, the five loudspeakers/channels are designated L, C, R, SL and SR. For the 4/4 presentation, the loudspeakers/channels are designated L, CL, CR, R, SL1, SL2, SR1 and SR2. Lesser arrangements are obvious reductions from the above.

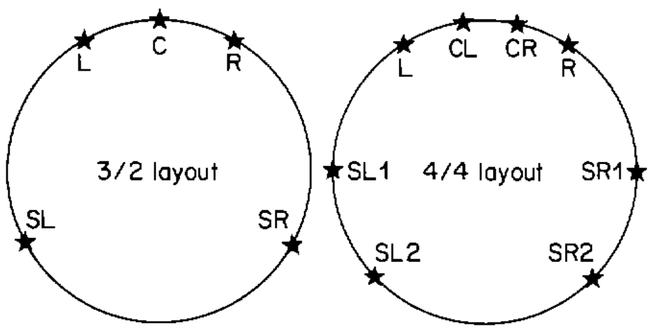

Fig. A1.1 - Loudspeaker layouts for surround sound.

A1.2 Downwards compatibility

Concentrating first on the downward compatibility, a mathematical matrix provides for simple reception in the lower orders of the hierarchy when higher orders of programme are being broadcast. For instance, if a 3/2 format programme is being broadcast the stereo receiver should only need to decode two transmission channels in order to drive the stereo loudspeakers.

The same applies to nearly all combinations of transmission and receive mode. A simplification can,

however, be made if there is a distinction between the auditoria formats of source and reception (seen here as 4/4, 4/2, and 3/4) and the domestic formats (seen here as 3/2, 3/1, 3/0, etc.). Undoubtedly, there is a need to be able to derive 3/2 transmission signals from, say, 4/4 production signals; but this derivation is considered a separate task from the transmission compatibility matrixing.

We can also deal separately with the front production channels and the surround production channels, as any reduction will be for either one group of channels or the other. Thus, dealing first with the front channels, there is a need to derive three intermediate signals (L', R', C') from the four production signals (L, R, CL, CR). Two proposals have been considered, namely simple addition of CL and CR such that:

$$\begin{aligned} C' &= CL + CR \\ L' &= L \\ R' &= R \end{aligned}$$

and more complex reduction such that:

$$\begin{aligned} C' &= .866*CL + .886*CR \\ L' &= L + .500*CL \\ R' &= R + .500*CR \end{aligned}$$

Preliminary tests on these two proposals have demonstrated a significant shortcoming of the simple proposal, in that any sound sources spread between CL and CR are collapsed, in the reduced format, to come from a single location: thus, the distribution of static sound sources or the evenness of movement across the front sound stage, are adversely affected. On this basis, the more complex reduction is recommended for further study.

Dealing secondly with the surround channels, there is no single obviously correct way of reducing from four channels (SL1, SL2, SR1, SR2) to two (SL, SR), as the requirements vary, depending on the contents of the four original surround signals. If these signals are four discrete sources, then simple addition is enough, such that:

$$\begin{aligned} SL &= .707*(SL1 + SL2) \\ SR &= .707*(SR1 + SR2) \end{aligned}$$

However, if, as has been suggested for some programmes, SL2 and SR2 have already been derived artificially from SL1 and SR1, then the reduction should only take account of the true surround material, such that:

$$\begin{aligned} SL &= SL1 \\ SR &= SR1 \end{aligned}$$

Thirdly, if the four source surround channels are derived from a coincident microphone arrangement, it may be necessary to use a weighted sum of the source channels:

$$\begin{aligned} SL &= k1*SL1 + k2*SL2 + k3*SR2 \\ SR &= k1*SR1 + k2*SR2 + k3*SL2 \end{aligned}$$

where k1, k2 and k3 are the weighting functions. The correct form of reduction will, therefore, depend on factors relating to the original signals; but experience may lead to a single reduction algorithm being found to be adequate.

Having thus reduced the higher levels of the hierarchy to the 3/2 format, a compatibility matrix can be derived that achieves the aims already reported. It was with these aims in mind that the audio encode and decode matrices given in Table A1.1 were derived. They comprise a group of equations that take the source (production) signals L, R, C, SL and SR, as they might come from a tape recorder, and combine them into five signals A, B, T, Q1 and Q2, for conveyance to the transmission encoding/modulation circuitry for ultimate broadcasting. These encoding matrices provide the essence of compatible reception, as well as providing a reversible matrix for the surround listener. The decode equations are subdivided according to the style of the listening equipment. They provide for listening in mono (1/0), stereo (2/0), three channel (3/0), four channel (3/1) and five channel (3/2).

Table A1.1: Five channel surround: encoding and decoding equations.

Encoding equations

		L	R	C	SL	SR
A	=	1.000	.000	.707	.707	.000
B	=	.000	1.000	.707	.000	.707
T	=	.000	.000	.707	.000	.000
Q1	=	.000	.000	.000	.707	.707
Q2	=	.000	.000	.000	.707	−.707

Decoding equations

Mono — 1/0 format

		A	B	T	Q1	Q2		L	R	C	SL	SR
M	=	.707	.707	.000	.000	.000	=	.707	.707	1.000	.500	.500

Stereo — 2/0 format

		A	B	T	Q1	Q2		L	R	C	SL	SR
L′	=	1.000	.000	.000	.000	.000	=	1.000	.000	.707	.707	.000
R′	=	.000	1.000	.000	.000	.000	=	.000	1.000	.707	.000	.707

Three channels — 3/0 format

		A	B	T	Q1	Q2		L	R	C	SL	SR
L′	=	1.000	.000	−1.000	.000	.000	=	1.000	.000	.000	.707	.000
R′	=	.000	1.000	−1.000	.000	.000	=	.000	1.000	.000	.000	.707
C′	=	.000	.000	1.414	.000	.000	=	.000	.000	1.000	.000	.000

Four channels — 3/1 format

		A	B	T	Q1	Q2		L	R	C	SL	SR
L′	=	1.000	.000	−1.000	−.500	.000	=	1.000	.000	.000	.354	−.354
R′	=	.000	1.000	−1.000	−.500	.000	=	.000	1.000	.000	−.354	.354
C′	=	.000	.000	1.414	.000	.000	=	.000	.000	1.000	.000	.000
S′	=	.000	.000	.000	1.000	.000	=	.000	.000	.000	.707	.707

Five channels — 3/2 format

		A	B	T	Q1	Q2		L	R	C	SL	SR
L′	=	1.000	.000	−1.000	−.500	−.500	=	1.000	.000	.000	.000	.000
R′	=	.000	1.000	−1.000	−.500	.500	=	.000	1.000	.000	.000	.000
C′	=	.000	.000	1.414	.000	.000	=	.000	.000	1.000	.000	.000
SL′	=	.000	.000	.000	.707	.707	=	.000	.000	.000	1.000	.000
SR′	=	.000	.000	.000	.707	−.707	=	.000	.000	.000	.000	1.000

Looking at the encode matrix, it has been arranged in tabular form. Thus the first line represents the equation:

$$A = 1.000*L + 0.000*R + 0.707*C + 0.707*SL + 0.000*SR$$

The decode matrix is similarly arranged, except that, having presented the appropriate equations for combining the transmitted signals A, B, T, Q1 and Q2, substitution is then made in order to show what the result is in terms of the original source signals L, R, C, SL and SR.

One aspect of the encode and decode matrix that becomes apparent with further analysis, is that, in some modes of reproduction, there is a variation in reproduced sound power level depending on source location. It should be recollected, however, that the same already applies to the M and S matrixing in stereo, which gives rise to a 3 dB variation. In this case, the variation is given in Fig. A1.2 for the specific case of the equations of Table A1.1 and sine/cosine panning of the sound source around the surround stage. The figure shows, for a five channel source (3/2 format), the sound power variation for reproduction in mono (1/0), stereo (2/0), three (3/0) and four (3/1) channel formats, for the different directions of centre front, left front, left surround and the intermediate locations. The decoding to five channels (3/2) is perfect, in that there is no variation of sound power level.

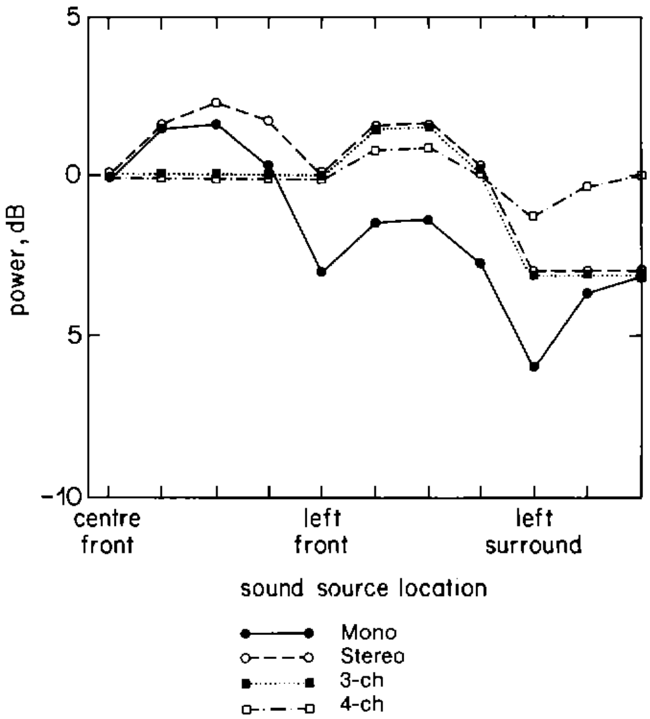

Fig. A1.2 - Audio compatibility matrix. Reproduced power variation: 5-ch surround sound.

APPENDIX 2

Downward Mixing of Multichannel Audio Signals

A2.1 Introduction

When stereo radio was being defined it was recognised that provision had to be made for the compatible reception of the signals in both mono and stereo. Thus, it was decided to transmit the signals:

$$M = (L+R)/2 \qquad \text{and} \qquad S = (L-R)/2$$

such, that simple mono reception could continue, whilst stereo was provided by the two signals and some simple decoding. The same is true today, with multichannel sound for either television or radio, that several forms of reception have to be accommodated, and thus compatibility matrixing has been proposed.

This assumes that there is a reason, such as compatibility with existing transmission formats or cheaper receivers, for providing the matrix at the transmission end of the broadcast chain. In the context of some of the bit-rate reduction proposals now being developed, it also has the drawback of requiring sum and difference matrixing at the receiver to re-establish the loudspeaker feed signals. This can, under some circumstances, expose the limitations of the bit-rate reduction systems, and hence fresh proposals are being tabled in CCIR and EBU working parties. These proposals suggest the transmission of loudspeaker feed signals for the highest member of the sound hierarchy, say 3/2 surround sound, with proportional summation being used to generate loudspeaker signals for lower members of the hierarchy, say a 3/0 presentation. This proportional summation is termed 'downward mixing'.

A2.2 The proposal

The downward mixing equations are presented in two tables, which refer respectively to 3/2 source material and 4/4 source material. They are arranged, in tabulated matrix form, to show reproduction equations for a number of loudspeaker arrangements.

Table A2.1 gives the downward mixing equations for 3/2 source material, based on the equations used for compatibility matrixing. It shows reproduction in formats such as mono (1/0), stereo (2/0), 3/0, 3/1 and 2/2. It should be remembered that its derivation was based to some extent on the subject appraisal of 3/2 material in such formats. It is relevant to note, that there is deliberate attenuation of the surround source channels in formats with reduced numbers of loudspeakers, to reduce the level of ambient sounds in the resulting mixes. Thus, in mono, the surround sources are attenuated by 6 dB, and in stereo, 3/0, 2/1 and 3/1 formats by 3 dB. Also, the surround information is panned to optimise the subjective impression given.

The downward mixing equations for 4/4 source material are shown in Table A2.2 for mono, stereo, 3/0, 4/0, 3/2, and 4/2 presentations etc. Their derivation follows the principles used for the 3/2 source material above. Again, there is attenuation of the surround sources in some of the reproduction modes. These are 6 dB in mono, stereo, 3/0 and 4/0 format; 3 dB in 2/1 and 3/1 format; 0 dB in 2/2, 3/2, and 4/2 formats. The panning of signals has been derived on the basis that the phantom images for loudspeaker signals in, say, stereo should be coincident with the 'missing' loudspeakers. Thus, in stereo, CL and CR sources are panned to positions of ±10 degrees from centre front. Other panned sources are derived on the same basis.

Table A2.1: Downward mixing equations for 3/2 source material.

Mono — 1/0 format

		L	R	C	SL	SR
C'	=	.707	.707	1.000	.500	.500

Stereo — 2/0 format

		L	R	C	SL	SR
L'	=	1.000	.000	.707	.707	.000
R'	=	.000	1.000	.707	.000	.707

Three channels — 3/0 format

		L	R	C	SL	SR
L'	=	1.000	.000	.000	.707	.000
R'	=	.000	1.000	.000	.000	.707
C'	=	.000	.000	1.000	.000	.000

Three channels — 2/1 format

		L	R	C	SL	SR
L'	=	1.000	.000	.707	.000	.000
R'	=	.000	1.000	.707	.000	.000
S'	=	.000	.000	.000	.707	.707

Four channels — 3/1 format

		L	R	C	SL	SR
L'	=	1.000	.000	.000	.000	.000
R'	=	.000	1.000	.000	.000	.000
C'	=	.000	.000	1.000	.000	.000
S'	=	.000	.000	.000	.707	.707

Four channels — 2/2 format

		L	R	C	SL	SR
L'	=	1.000	.000	.707	.000	.000
R'	=	.000	1.000	.707	.000	.000
SL'	=	.000	.000	.000	1.000	.000
SR'	=	.000	.000	.000	.000	1.000

Table A2.2: Downward mixing equations for 4/4 source material

Mono — 1/0 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
C′	=	.707	.707	.707	.707	.354	.354	.354	.354

Stereo — 2/0 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.866	.500	.500	.000	.433	.250
R′	=	.000	1.000	.500	.866	.000	.500	.250	.433

Three channels — 3/0 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.500	.000	.500	.000	.250	.000
R′	=	.000	1.000	.000	.500	.000	.500	.000	.250
C′	=	.000	.000	.866	.866	.000	.000	.433	.433

Three channels — 2/1 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.866	.500	.000	.000	.000	.000
R′	=	.000	1.000	.500	.866	.000	.000	.000	.000
S′	=	.000	.000	.000	.000	.707	.707	.707	.707

Four channels — 4/0 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.000	.000	.500	.000	.000	.000
R′	=	.000	1.000	.000	.000	.000	.500	.000	.000
CL′	=	.000	.000	1.000	.000	.000	.000	.500	.000
CR′	=	.000	.000	.000	1.000	.000	.000	.000	.500

Four channels — 2/2 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.866	.500	.000	.000	.000	.000
R′	=	.000	1.000	.500	.866	.000	.000	.000	.000
SL′	=	.000	.000	.000	.000	1.000	.000	.866	.500
SR′	=	.000	.000	.000	.000	.000	1.000	.500	.866

Four channels — 3/1 format

		L	R	CL	CR	SL1	SR1	SL2	SR2
L′	=	1.000	.000	.500	.000	.000	.000	.000	.000
R′	=	.000	1.000	.000	.500	.000	.000	.000	.000
C′	=	.000	.000	.866	.866	.000	.000	.000	.000
S′	=	.000	.000	.000	.000	.707	.707	.707	.707

Table A2.2: Downward mixing equations for 4/4 source material (cont.).

Five channels — 3/2 format

	L	R	CL	CR	SL1	SR1	SL2	SR2
L' =	1.000	.000	.500	.000	.000	.000	.000	.000
R' =	.000	1.000	.000	.500	.000	.000	.000	.000
C' =	.000	.000	.866	.866	.000	.000	.000	.000
SL' =	.000	.000	.000	.000	1.000	.000	.866	.500
SR' =	.000	.000	.000	.000	.000	1.000	.500	.866

Six channels — 4/2 format

	L	R	CL	CR	SL1	SR1	SL2	SR2
L' =	1.000	.000	.000	.000	.000	.000	.000	.000
R' =	.000	1.000	.000	.000	.000	.000	.000	.000
CL' =	.000	.000	1.000	.000	.000	.000	.000	.000
CR' =	.000	.000	.000	1.000	.000	.000	.000	.000
SL' =	.000	.000	.000	.000	1.000	.000	.866	.500
SR' =	.000	.000	.000	.000	.000	1.000	.500	.866

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